User Manual

GL-VP-150

IP Phone

http://www.giga-link.ru

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1. WELCOME

The IP PHONE GL-VP-150 is an internet telephone set that features superb audio quality, rich functionalities, high level of integration, and compactness. By converting analog voice for transmission over the internet, the IP Phone GL-VP-150 allows users with broadband internet connections to make calls to and from anywhere in the world. The IP PHONE GL-VP-150 is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market

2. WHAT IS IN THE PACKAGE

The GL-VP-150 package contains:

- One GL-VP-150 VoIP Phone
- One universal power supply
- One Ethernet cable

3. PRODUCT OVERVIEW

4. Key Features

- Support SIP 2.0 (RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP/RARP, DNS, DHCP, NTP, PPPoE, STUN, UPNP,TFTP, etc.
- Powerful Digital Signal Processing (DSP) technology to ensure superior audio quality
- Support various codecs including G.711 (PCM a-law and u-law), G.723.1 , G.729A, G.726.
- Support standard encryption and authentication (DIGEST using MD5, MD5-sess)
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Acoustic Echo Cancellation (AEC) with Acoustic Gain

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Control(AGC) for speakerphone mode.

- Support automated provisioning for mass deployment ,RTP and TLS (pending)for security protection
- Support automated NAT traversal without manual manipulation of firewall/NAT
- Support numeric Caller ID Display, Hold, Transfer, Forward, 3-way Conference, in-band and out-of-band DTMF, Download Phone Book(XML, up to 120 items), Call Waiting, Call Log, Off-hook Auto Dial, Auto Answer, Downloadable Ringtones, SMS,Direct IP Call, Intercom, Paging, Pick up.
- Support syslog, full duplex hands-free speakerphone with advanced acoustic echo cancellation, redial, volume control, voice mail with indicator, downloadable ring tones.
- Provide easy configuration through manual operation (phone keypad), Web interface or automated centralized configuration file via TFTP or HTTP
- Support 5 navigation/menu/volume keys, 13 dedicated function keys for: HOLD, SPEAKERPHONE, MUTE, TRANSFER, CONFERENCE, and MESSAGE (with message indicator), 6 SPEED KEYS.
- Support multi-language: English, Chinese, Japan, etc

Feature Highlights

- 80×39mm LCD with back-lit，support 4 lines with 16 characters every line
- dual switched audio –sensing 10/100Mbps network ports.
- Support headset jack
- Support xml phone book, SMS etc.

4.1. Hardware specification

Model	GL-VP-150
LAN interface	1x RJ45 100Base-T
PC interface	1x RJ45 100Base-T
Universal switching power supply	Input: 100-240VAC 50-60 Hz Output: +5VDC, 1200mA CE/FCC/UL certification
Dimension	215 x 165 x 70 mm (L x W x H)
Weight	0.9kg
Temperature	40 – 130 F 5 – 45 C
Humidity	10 - 90%

5. INSTALLATIONS

Following are the steps to install a GL-VP-150 :

- Insert power adapter into back of the phone and connect it to a power outlet
- Connect internet cable from back of the phone (WAN Port) to broadband modem, router,

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hub, or switch

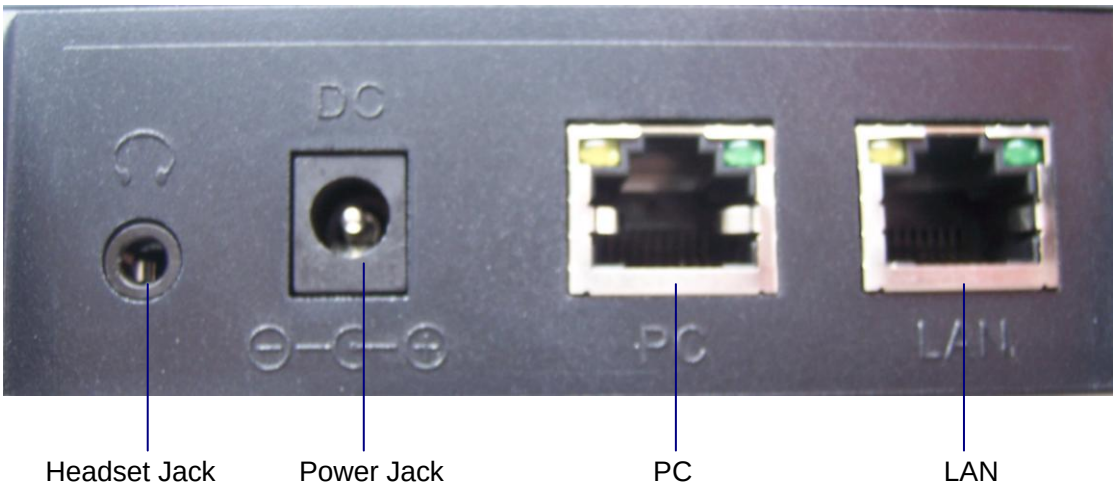
➢Connect internet cable from back of the phone (PC Port) to a PC.

SAFETY COMPLIANCES

The GL-VP-150 phone complies with FCC/CE and various safety standards. The GL-VP-150 power adaptor is compliant with the UL standard. Only use the universal power adaptor provided with the GL-VP-150 package. The manufacturer's warranty does not cover damages to the phone caused by unsupported power adaptors.

WARRANTY

If you purchased your GL-VP-150 from a reseller, please contact the company where you purchased your phone for replacement, repair or refund. If you purchased the product directly from GigaLink technologies, contact your GigaLink's Sales and Service Representative for a RMA (Return Materials Authorization) number before you return the product. GigaLink reserves the right to remedy warranty policy without prior notification.



Power Jack	5V DC power port; UL Certified
PC	10/100Mbps RJ-45 ports for PC (downlink) connection.
LAN	10/100Mbps RJ-45 port for LAN (uplink) connection.
Headset Jack	3.5mm

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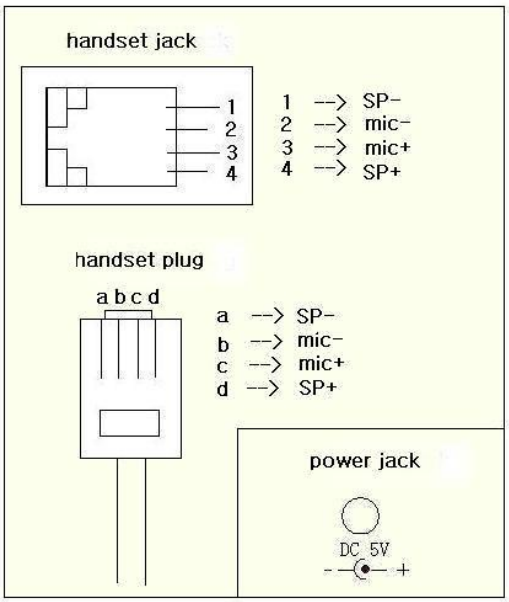
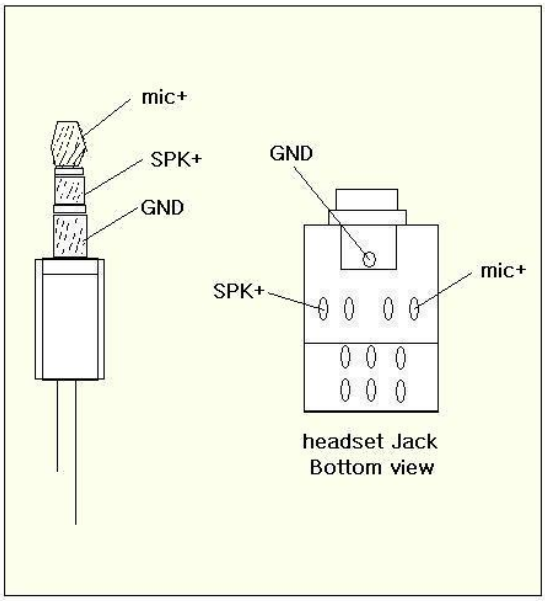
GL-VP-150 interface wiring schema:





Note: GL-VP-150 use 3.5mm headset.

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6. Get Familiar With The Buttons

6.1. Keypad Button

Kepad Buttons	Kepad Buttons Definitions
◄	Return to the last screen and shortcut key for phone book.
▲	Scroll to previous menu/submenu selection, or rise the volume of the loudhailer
✔	Enter Keypad Configuration “MENU” mode when phone is in IDLE mode. Use as ENTER key when in Keypad Configuration.
▼	Scroll to next menu/submenu selection, or lower the volume of the loudhailer
►	Go to the next screen and shortcut key for Dialed calls.
✉	Enter to retrieve voice mails or other messages

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0 - 9, *(.), #	To input: numbers, *(.), #
TRNF	Transfer the present call to another number
HOLD	Hold the present call for the moment
FLASH	Swith the call
🕮	Brings phonebook on screen
✕	Delete the call history, voice message, input keys, Reject Call,Mute,etc.
CONF	Conference call for three sides
# and SEND	Press SEND to dial a new number or redial the last number dialed. Press send button to send a call immediately before “no key entry timeout” value Expires
	Enable/Disable hands-free speaker mode
Speed key buttons	Short cut of register call

7. BASIC OPERATIONS

7.1. Make Phone call

7.1.1. Handset, Speakerphone and Headset Mode

Handset can be toggled between Speaker and Headset. To switch between Handset and Speaker/Headset, press the SPEAKER button.

7.1.2. Completing Calls

There are four ways to complete a call:

- **DIAL:** To make a phone call.
  - • Take Handset/SPEAKER/Headset off-hook
  - • The phone will have a dial tone and the LCD display the “Dail”.
  - • Enter the phone number
  - • Press the SEND key
- **REDIAL:** To redial the last dialed phone number.

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When redialing the phone will use the same SIP account as was used for the last call.

• Take Handset/SPEAKER/Headset off-hook or

• Press the # or SEND button

➢ **USING THE CALL HISTORY:** To call the a phone number in the phone's history

When using the call history, the phone will use the same SIP account as was used for the last call/call attempt. Thus, when returning a call made to the third SIP account, the phone will use the third SIP account return the call.

•Press the MENU button to bring up the Main Menu.

•Select Call History and then "Received Calls", "Missed Calls" or "Dialed Calls" depending on your needs

• Select phone number using the arrow keys

• Press OK to select

• Press OK again to dial.

➢ **USING THE PHONEBOOK:** Calling a phone in from the phone's phonebook.

• Go to the phonebook by:

. Pressing the left arrow button (at idle or talking status), or

Pressing the menu button and Selecting "Phone book" and Press MENU

• Select the phone number by using the arrow keys

• Press OK so select

• Press OK again to dial.

7.1.3. Making Calls using IP Addresses

Direct IP calling allows two phones to talk to each other in an ad hoc fashion without a SIP proxy. VoIP calls can be made between two phones if:
• Both phones have public IP addresses, or
• Both phones are on a same LAN/VPN using private or public IP addresses, or
• Both phones can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ)
To make a direct IP call, please follow these steps:
1. Press MENU button to bring up MAIN MENU.
2. Select "Direct IP Call" using the arrow-keys.
3. Press OK to select.

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4. Input the 12-digit target IP address. (Please see example below).
5. Press OK key to initiate call.

To make a quick IP call, please see next section.

For example: If the target IP address is 192.168.1.60 and the port is 5062 (e.g. 192.168.1.60:5062), input the following: **192*168*1*60*5062 -** The " ***** " key represent the dot"**.**" And colon "**:**".

Press OK to dial out

7.1.4. Quick IP Call Mode

The GL-VP-150 also supports Quick IP call mode. This enables the phone to make direct IP-calls, using only the last few digits (last octet) of the target phone's IP-number.
This is possible only if both phones are in under the same LAN/VPN. This simulates a PBX function using the CMSA/CD without a SIP server. Controlled static IP usage is recommended.

For example:

192.168.0.2 calling 192.168.0.3 -- dial *473 follow by SEND or #
192.168.0.2 calling 192.168.0.23 -- dial *4723 follow by SEND or #
192.168.0.2 calling 192.168.0.123 -- dial *47123 follow by SEND or #
192.168.0.2: dial *473 and *4703 and *47003 results in the same call -- call 192.168.0.3

NOTE: If you have a SIP Server configured, a Direct IP-IP still works. If you are using STUN, the Direct IPIP call will also use STUN. Configure the "Use Random Port" to "NO" when completing Direct IP calls.

7.2. ANSWERING PHONE CALLS

7.2.1. Receiving Calls

1. Incoming single call: Phone rings with selected ring-tone. The LCD will display the caller's. Answer call by taking Handset/SPEAKER/Headset off hook or pressing SPEAKER.

2. Incoming multiple calls: When another call comes in while having an active call, the phone will produce a Call Waiting tone (stutter tone). Answer the incoming call by pressing the FLASH key. The current active call will be put on hold.

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7.3. PHONE FUNCTIONS DURING A PHONE CALL

7.3.1. Call Hold

While in conversation, pressing the “Hold” button will put the remote end on hold. Pressing the “Hold” button again will release the previously Hold state and resume the bi-directional media.

7.3.2. Call Waiting and Call Flashing

If call waiting feature is enabled, while the user is in a conversation, he will hear a special stutter tone if there is another incoming call. User then can press FLASH button to put the current call party on hold automatically and switch to the other call. Pressing flash button toggles between two active calls.

7.3.3. Call Transfer

GL-VP-150 supports both blind and attended call transfer. Each is easy to use. Use blind transfer if you want to transfer a call without speaking with someone first; use attended transfer if you want to speak with the someone prior to transferring call.

7.3.3.1 Blind Transfer

Transfer an active call to a third party without announcement.

Press the TRANSFER button and wait for a dial tone. Dial the third party’s phone number followed by the SEND button.

NOTE: The “Enable Call Feature” must be configured to “Yes” in the web configuration page to enable this feature.

7.3.3.2 Attended Transfer

Transfer an active call to a third party with attended.

Press Flash button and make a call and automatically place the ACTIVE call on HOLD. Once the call is established, hang up to transfer the call.

NOTE: To transfer calls across SIP domains, SIP service providers must support transfer across SIP domains. Blind transfer will usually use the primary account.

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7.3.4. Conference Call

GL-VP-150 phone supports 3-way conference.

Assuming that call party A and B are in conversation. A wants to bring C in a conference:

- A presses the “CONFERENCE” button to get a dial tone and put B on hold
- A dials C’s number then “SEND” key to make the call
- If C answers the call, then A presses “CONFERENCE” button to bring B, C in the conference.
- If C does not answer the call, A can press FLASH back to talk to B.

NOTE:

- During the conference, if B or C drops the call, the remaining two parties can still talk. However, if A the conference initiator hangs up, all calls will be terminated.

7.3.5. Mute/Delete/Reject incoming calls

1. Press the ✕ button to enable/disable muting the microphone.
2. The “Line Status Indicator” will show “LINEx: SPEAKING” or “LINEx: MUTE” to indicate whether the microphone is muted.

NOTE: Pressing ✕ button for an incoming call will reject the call. ✕ button also functions as delete key when user wish to delete the last entered digit.

7.3.6. Voice Messages (Message Waiting Indicator)

A blinking red MWI (Message Waiting Indicator) indicates a message is waiting. Press the MSG button to retrieve the message. An IVR will prompt the user through the process of message retrieval.

NOTE: Account requires a voicemail portal number to be configured in the “voicemail user id” field.

7.3.7. SMS (Short Messaging Service)

The sms can only be used if the SERVER/PBX supports this feature.

If the phone has received a sms, the LCD screen will display the NEW IM.

Press the MENU key, and select the Instand Message, and press OK, the you can check the Input message.

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7.3.8. Do Not Disturb

The “DND” functinon can be set at config menu to set phone to ‘do not disturb’ (it will be on the screen). The phone will not ring and send caller directly to voicemail.

7.4. CALL FEATURES

7.4.1. Call Features Tables

Following table shows the call features of GL-VP-150 :

Key	Call Features
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)
*67	Block Caller ID (per call)
*82	Send Caller ID (per call)
*50	Disable Call Waiting (for all subsequent calls)
*51	Enable Call Waiting (for all subsequent calls)
*70	Disable Call Waiting. (Per Call)
*71	Enable Call Waiting (Per Call)
*72	Unconditional Call Forward. To use this feature, dial “*72” and get the dial tone. Then dial the forward number and “#” for a dial tone, then hang up.
*73	Cancel Unconditional Call Forward. To cancel “Unconditional Call Forward”, dial “*73” and get the dial tone, then hang up.
*90	Busy Call Forward. To use this feature, dial “*90” and get the dial tone. Then dial the forward number and “#” for a dial tone, then hang up.
*91	Cancel Busy Call Forward. To cancel “Busy Call Forward”, dial “*91” and get the dial tone, then hang up.
*92	Delayed Call Forward. To use this feature, dial “*92” and get the dial tone. Dial the forward number and “#” for a dial tone and then hang up.
*93	Cancel Delayed Call Forward. To cancel this feature, dial “*93”, get the dial tone, and then hang up.
Flash/Hook	call waiting indication. When in conversation without an incoming call, this action will switch to a new channel to make a new call.

7.4.2. XML Phonebook

The GL-VP-150 supports a download XML phonebook.

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8. CONFIGURATION GUIDE

8.1. CONFIGURATION VIA KEYPAD

To enter the MENU, press the central button. Navigate the menu by using the arrow keys: up/down and left/right.

Call History	Displays histories of incoming, dialed and missed calls.
Phone Book	Displays the New Entry and you can download the phonebook.
Status	Displays the network status, account statuses, software version and MAC-address of the phone.
Instant Message	Goes to instant messages
Direct IP Call	Displays the ip call options menu
Config	Press *Menu* button to display the configuration selections: • **Network** To enable/disable DHCP. To setup IP-address, Net mask and Gateway address • **Account** • **Upgrade** In this menu setting regarding the firmware server and configure server can be changed. It also enables the user to make the phone attempt to download new firmware. • **Factory Reset** Press *Menu* button and the Confirm Factory Reset to reset FACTORY DEFAULT setting. Do not use Factory Reset unless you want to restore factory settings Press '◀' to return the main menu.
Preference	Press *Menu* button to enter this sub menu including • **Ring Tone** Choose different ring tones in the "Ring Tone" menu.

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	• **Ring Volume** Press Menu button to hear the selected ring volume, press '▲' or '▼' to hear and adjust the ring tone volume.. Press '◀' to return to the main menu.
Factory Functions	Press *Menu* to display the factory function items including • **Audio Loopback** Speak into the handset. If you hear your voice in the handset, your audio works fine. • **Diagnostic Mode** All LEDs will light up Press any key on the keypad, to display the button name in the LCD. Lift and put back the handset. Press '◀' to return to the main menu.
Reboot	Press *Menu* button to reboot the device And choose Confirm Reboot or Cancel Reboot.

The main configuration menu is the *Config* menu. This menu consists of four submenus that allow you to configure and display information about the Unicore5001. Table 5 shows the available menus and their uses.

Configuration Menus and Their Functions

option	Used To
Network	assign network settings such as the *IP Mode*, IP, Netmask and Gateway
Account	assign Account information such as the SIP Proxy, Outbound Proxy, User ID, Auth ID, Password, User Name, Audio, Confirm Save, Return & Cancel.
Upgrade	assign Upgrade Settings such as the Firmware Server, Configure Server, Upgrade Via.
Factory Reset	Choose whether to restore factory settings

Navigating the Menus

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The following information will help you understand the interface for the configuration menus.

- Press the **MENU** key to access the *Setup* menu.
- Use the ▲ and ▼ keys to scroll through the menu choices.
- When the desired menu option is indicated by →, press the **MENU** key again to select the indicated option.
- Press the ◀◀ to return to the previous screen.

Network Setting

The ***Network Setting*** sub-menu is used to assign network settings such as the ***IP Mode***, **IP**, ***Netmask*** and ***Gateway***. All sub-menus under *Network Setting* are password protected and can be altered only by an administrator. Users may view the information but not change it.

Step1: Press the **MENU** key to enter the *Setup* menu

Step2: Choose the *Config* sub-menu and press **MENU** again.

Step3: Choose the *Network* sub-menu and press **MENU**

Step4: Chooose the *IP Mode* sub-menu and press **MENU**

Step5: Use the ▼ and ▲ keys to choose the mode you want

Then press ✔ to go to the sub-menu *IP Mode,* and config IP, Netmask, Gateway as above.

IP display the IP address at the moment, you can input new IP address while the DHCP is set OFF.(Press **MENU** key to return to the previous screen.)

Account Setting

The Account Setting sub-menu is used to assign Account information such as the ***SIP Proxy, Outbound Proxy, User ID, Auth ID, Password, User Name, Audio, Confirm Save*** and ***Return & Cancel***.

Step1: Return to the Config sub-menu

Step2: Choose the *Acct* sub-menu and press **MENU** again.

Step3: Choose the SIP Proxy sub-menu and press **MENU** again.

Step4: Input the valid SIP proxy server and press **MENU** key to return to the previous screen.

You can config Outbound Proxy, User ID, Auth ID, Password, User Name, Audio as

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above.(the specifications of these can be found from 8.2.4.4 Account Options) and choose Confirm Save or Return & Cancel after you finish configuration.

Upgrade Setting

The ***Upgrade Setting*** sub-menu is used to assign Upgrade Settings such as the ***Firmware Server, Configure Server*** and ***Upgrade Via.***

Step1: Return to the Config sub-menu.

Step2: Choose the Upgrade sub-menu and press **MENU** again.

Step3: Choose the Firmware Server sub-menu and press **MENU** again.

Step4: Input the valid Firmware Server and press **MENU** key to return to the previous screen

You can config Configure Server, Upgrade Via as above. (the specifications of these can be found from 8.2.4.3 Super Options)

Factory Reset Setting

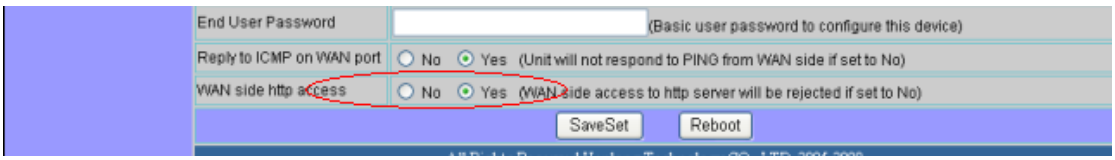
Step1: Return to the Config sub-menu.

Step2: Choose the Factory Reset sub-menu and press **MENU** again.

Step3:Choose Confirm Factory Reset or Cancel Factory Reset.

8.2. Configuring GL-VP-150 using Web Browser

GL-VP-150 has embedded Web server and HTML pages that allow users to configure the GL-VP-150 through an easy-to-use Web browser interface such as Microsoft's Internet Explorer or Netscape browser. Below is a screen shot of the GL-VP-150 configuration page:


End User Password
(Basic user password to configure this device)
Reply to ICMP on WAN port
No
Yes
(Unit will not respond to PING from WAN side if set to No)
WAN side http access
No
Yes
(WAN side access to http server will be rejected if set to No)
SaveSet
Reboot

8.2.1. Accessing the Web Configuration

The GL-VP-150 configuration page can be accessed via your web browser by entering the WAN IP address:

http://192.168.0.13

Be sure that your PC is connected to the router/hub/switch directly .

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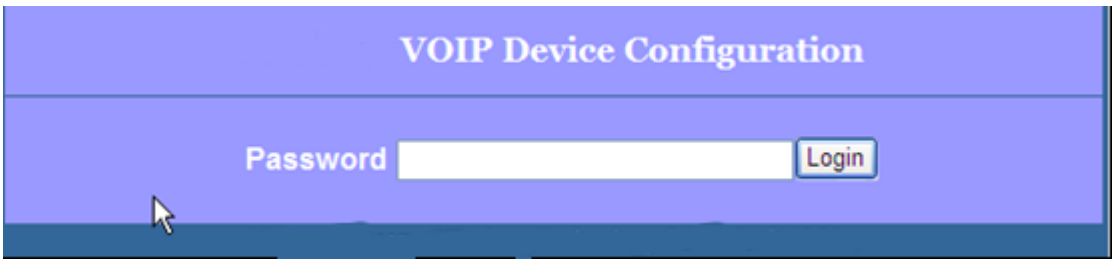
8.2.2. User Programming and Configuration

From your web browser, the GL-VP-150 will show the following login screen:



Enter the password and click on the “Login” button

8.2.3. Passwords

Passwords are case sensitive and all GL-VP devices come with factory default passwords as indicated below:

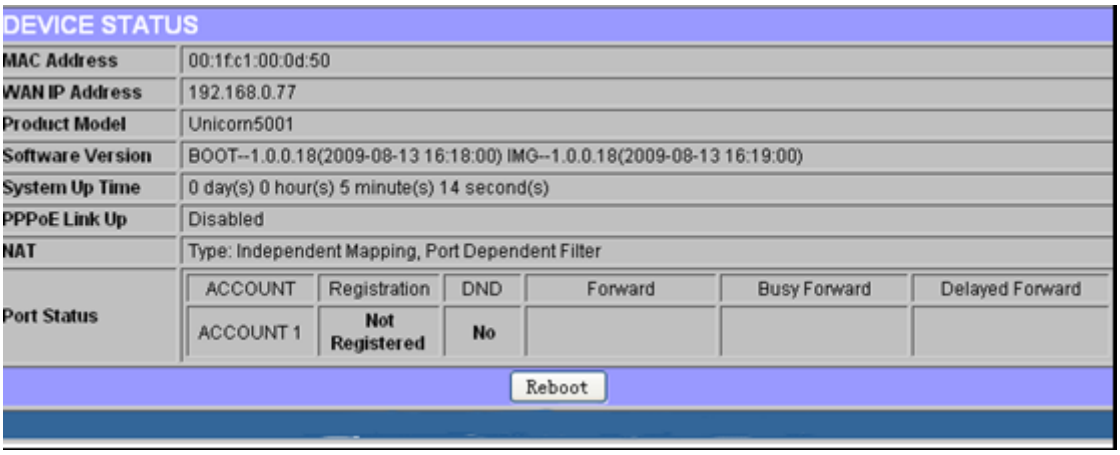
Advanced User Password for access to Super User Options: admin

End User Password for access to Basic User Options: 1234

8.2.4. Configuration Options and Explanations

After a correct password is entered in the login screen, the embedded web server inside the GL-VP-150 will show the configuration page, which is explained in details below:

8.2.4.1 Device Status



Device Status

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<table>
<tr><th>Options</th><th>Meaning</th></tr>
<tr><td>MAC Address</td><td>The device ID, in HEX format.This is a very important ID for ISP troubleshooting.</td></tr>
<tr><td>WAN IP Address</td><td>This field shows IP address of GL-VP-150.</td></tr>
<tr><td>Product Model</td><td>Contains the product model info.</td></tr>
<tr><td>Software Version</td><td>.• Program: This is the main software (firmware) release number, always used to identify the software (firmware) system of the phone.
• Boot: Booting code version number</td></tr>
<tr><td>System Up Time</td><td>Shows system up time since the last reboot.</td></tr>
<tr><td>PPPoE Link Up</td><td>Indicates whether the PPPoE connection is enabled (connected to a modem).</td></tr>
<tr><td>NAT</td><td>Indicates the type of NAT connection used by the GL-VP-150 via its WAN port.
Based on STUN protocol.</td></tr>
<tr><td>Port Status</td><td>Shows several information regarding the individual FXS ports.
<table>
<tr><th>Account</th><th>Registration</th><th>DND</th><th>Forward</th><th>Busy Forward</th><th>Delayed Forward</th></tr>
<tr><td>Account 1</td><td>Registered</td><td>No</td><td></td><td></td><td></td></tr>
</table>
Indicates whether account are registered to the related SIP server(s).</td></tr>
</table>

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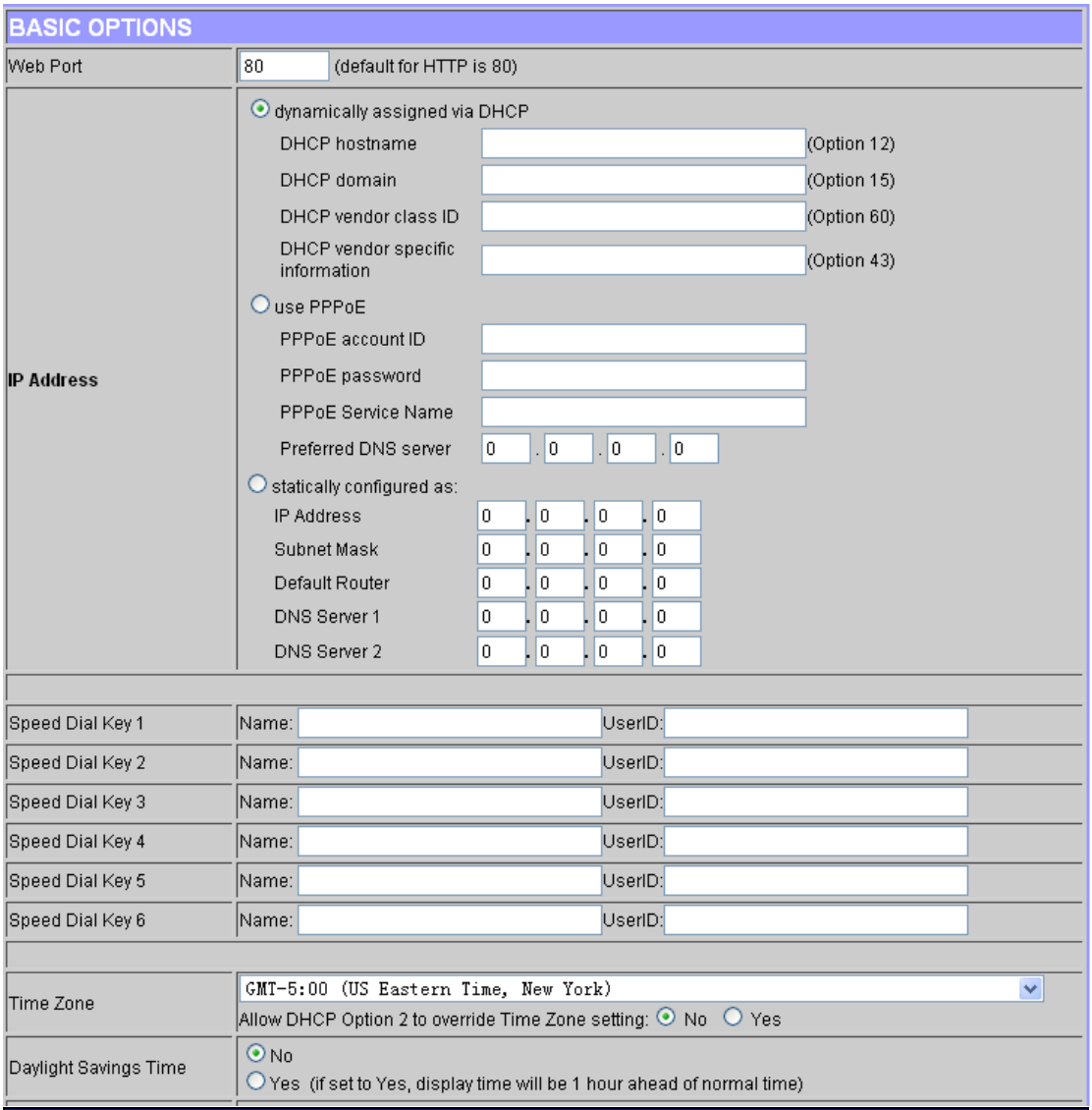
8.2.4.2 Basic Options



BASIC OPTIONS SETTING	
Options	**Meaning**
Web Port	Default is 80.
IP Address	There GL-VP-150 operates in two modes: 1. **DHCP mode**: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The GL-VP-150 acquires its IP address from the first DHCP server it discovers on its LAN. The DHCP option is reserved for NAT router mode. To use the PPPoE feature, set the PPPoE account settings. The GL-VP-150 establishes a PPPoE session if any of the PPPoE fields are set.

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	2. **Static IP mode**: configure all of the following fields: IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary). These fields are set to zero by default.
Speed Dail Key1~6	GL-VP-150 has define 6 speed dial keys. After you program numbers for this key. You can touch a speed dial key and then the call will be originated.
Time Zone	This parameter controls the date/time display according to the specified time zone.
Daylight Savings Time	This parameter controls time displayed in daylight savings time. If set to “Yes”, then the displayed time will be 1 hour ahead of normal time. The “Optional Rule” is configured to automatically adjust the Daylight Savings Time (DST) based on the rule set in this field. Rule Syntax: • start-time; end-time; saving • Both start-time and end-time have the same syntax: month, day, weekday, hour, minute o month: 1,2,3,..,12 (for Jan, Feb, .., Dec) o day: [+|-]1,2,3,..,31 o weekday: 1, 2, 3, .., 7 (for Mon, Tue, .., Sun), or 0 which means the daylight saving rule is not based on week days but based on the day of the month. o hour: hour (0-23), minute: minute (0-59) If “weekday” is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the “day” value must not be negative. If “weekday” is not zero and “day” is positive, then the daylight saving starts on the first “day” the iteration of the weekday (e.g.: 1st Sunday, 3rd Tuesday etc). If “weekday” is not zero and “day” is negative, then the daylight saving starts on the last “day” the iteration of the weekday (e.g.: last Sunday, 3rd last Tuesday etc). The saving is in the unit of minutes. The saving time may also be preceded by a negative (-) sign if subtraction is desired instead of addition. The default value is set for US, the “Automatic Daylight Saving Time Rule” shall be set to “3,2,7,2,0;11,1,7,2,0;60”

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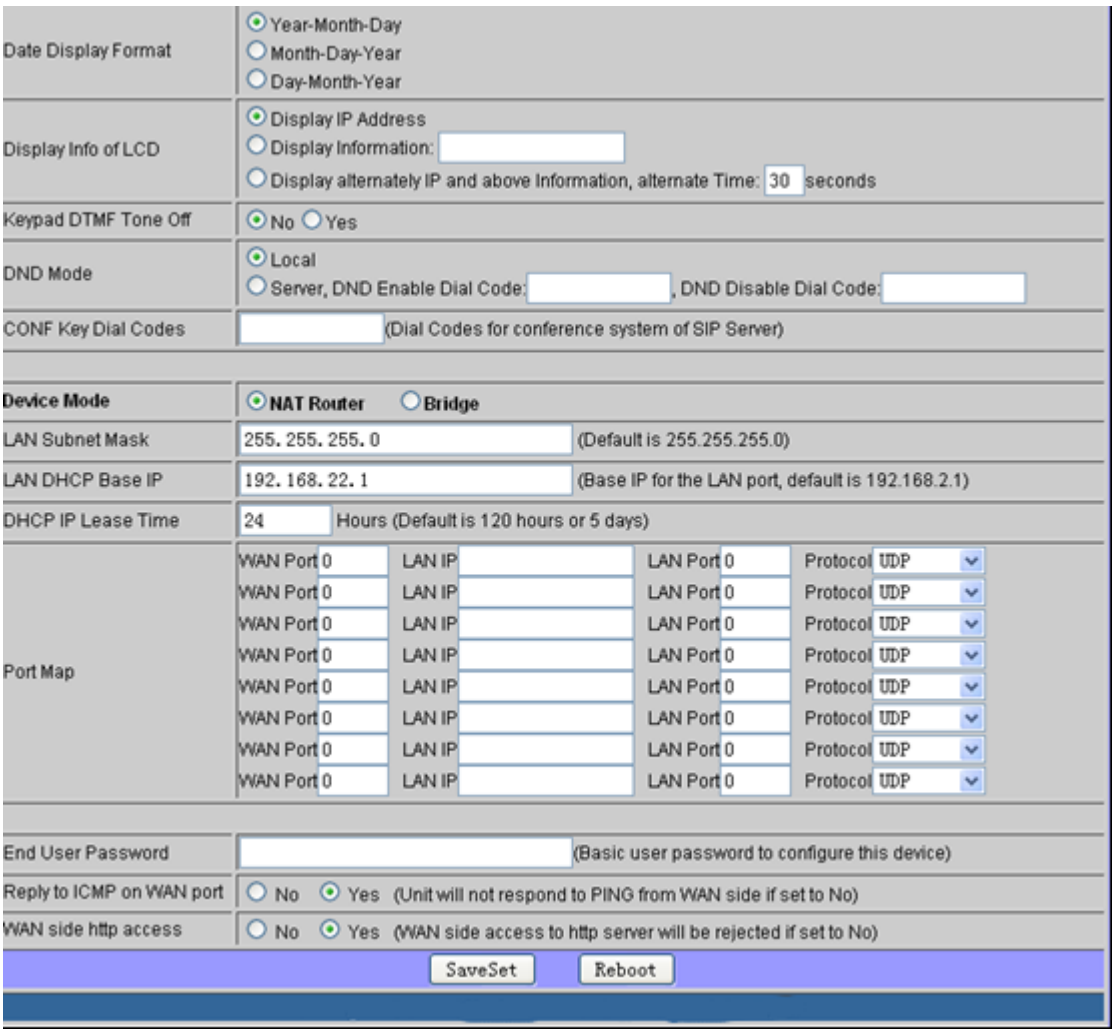
	Examples US/Canada where daylight saving time is applicable: 03,02,7,02,00;11,1,7,02,00;60 This means the daylight saving time starts from the second Sunday of March at 2AM and ends the first Sunday of November at 2AM. The saving is 60 minutes.


Date Display Format
Year-Month-Day
Month-Day-Year
Day-Month-Year
Display Info of LCD
Display IP Address
Display Information:
Display alternately IP and above Information, alternate Time: 30 seconds
Keypad DTMF Tone Off
No Yes
DND Mode
Local
Server, DND Enable Dial Code: , DND Disable Dial Code:
CONF Key Dial Codes
(Dial Codes for conference system of SIP Server)
Device Mode
NAT Router Bridge
LAN Subnet Mask
255. 255. 255. 0
(Default is 255.255.255.0)
LAN DHCP Base IP
192. 168. 22. 1
(Base IP for the LAN port, default is 192.168.2.1)
DHCP IP Lease Time
24
Hours (Default is 120 hours or 5 days)
Port Map
WAN Port 0 LAN IP LAN Port 0 Protocol UDP
End User Password
(Basic user password to configure this device)
Reply to ICMP on WAN port
No Yes (Unit will not respond to PING from WAN side if set to No)
WAN side http access
No Yes (WAN side access to http server will be rejected if set to No)
SaveSet
Reboot

BASIC OPTIONS SETTING	
Options	**Meaning**
Date Display Format	Allow user to choose among the following three formats: Year-Month-Day Month-Day-Year Day-Month-Year
Display info of LCD	You can select displaying ip address or you can custimize it.
Device Mode	To use the device as a router or a bridge.

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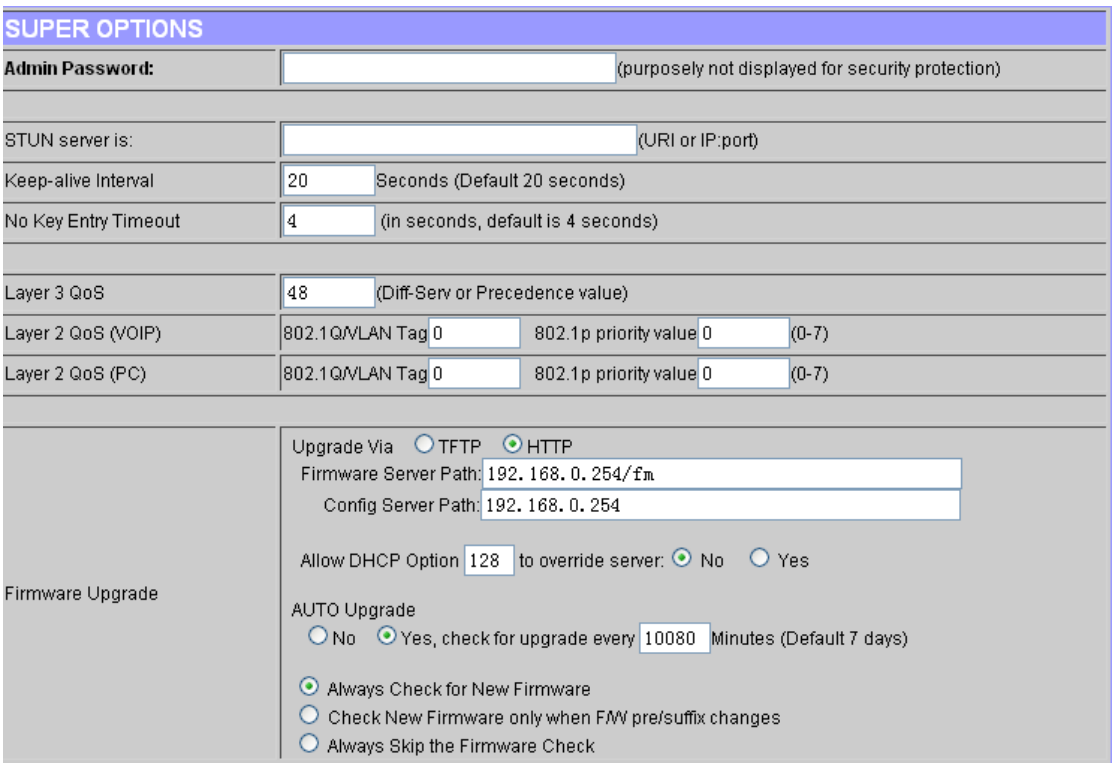
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0
LAN DHCP Base IP:	Base IP for the LAN port which functions as a Gateway for the subnet. Default value is 192.168.22.1
DHCP IP Lease Time:	Value is set in units of hours. Default value is 120hr (5 Days) The time IP address are assigned to the LAN clients
DMZ IP:	Forward all WAN IP traffic to a specific IP address if no matching port is used by GL-VP-150 itself or in the defined port forwarding
Port Map	Allow user to forward a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port.
End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive.
Reply to ICMP on WAN port	If set to "Yes", the GL-VP-150 will respond to the PING command from other computers, but it also is vulnerable to the DOS attack. Default is **No**.
Wan Side Http Access	If this parameter is set to "No", the HTML configuration update via WAN port is disabled.

8.2.4.3 Super Options



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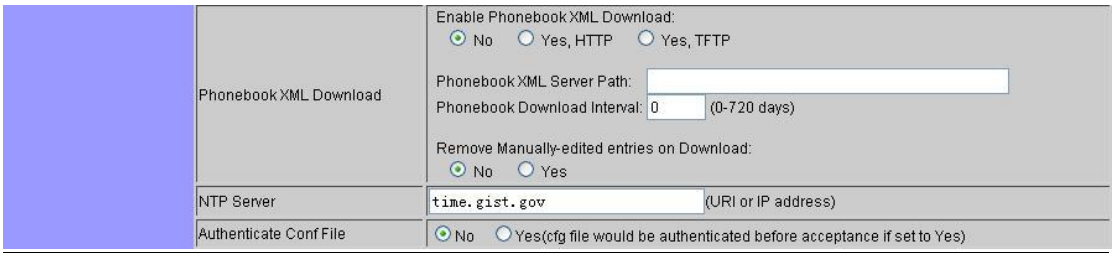

Super Options	
Setting Options	Meaning
Admin Password	Administrator password. Only the administrator can access the “Super Options” and “Account Settings” page. Password field is purposely blank for security reasons after clicking update and saved. The maximum password length is 25 characters.
STUN server is：	IP address or Domain name of the STUN server.
Keep-alive interval	This parameter specifies how often the GL-VP-150 sends a blank UDP packet to the SIP server in order to keep the “hole” on the NAT open. Default is 20 seconds. Minimum value is 20 seconds.
No Key Entry Timeout	Default is 4 seconds.
Layer3 Qos	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is **48**.
Layer2 Qos（VOIP）	Value used for layer 2 VLAN tag. Default setting is **blank**
Layer2 Qos （PC）	Layer 2 QoS settings for LAN port device traffic. Default setting is blank. VLAN supported equipment is required if user needs to change these settings.
Firmware Upgrade	Support firmware upgrade via TFTP or HTTP, Support for Authenticating configuration file before accepting changes User specific URL for configuration file and firmware files
Phonebook XML	Enable the XML phonebook via TFTP or HTTP. Define XML server path and download speed.
NTP server	URI or IP address of the NTP (Network Time Protocol) server. Used by the phone to synchronize the date and time.
Authenticate Conf File	Default is “No”. If set to “Yes”, configuration file would be authenticated before acceptance. End user should use default setting.

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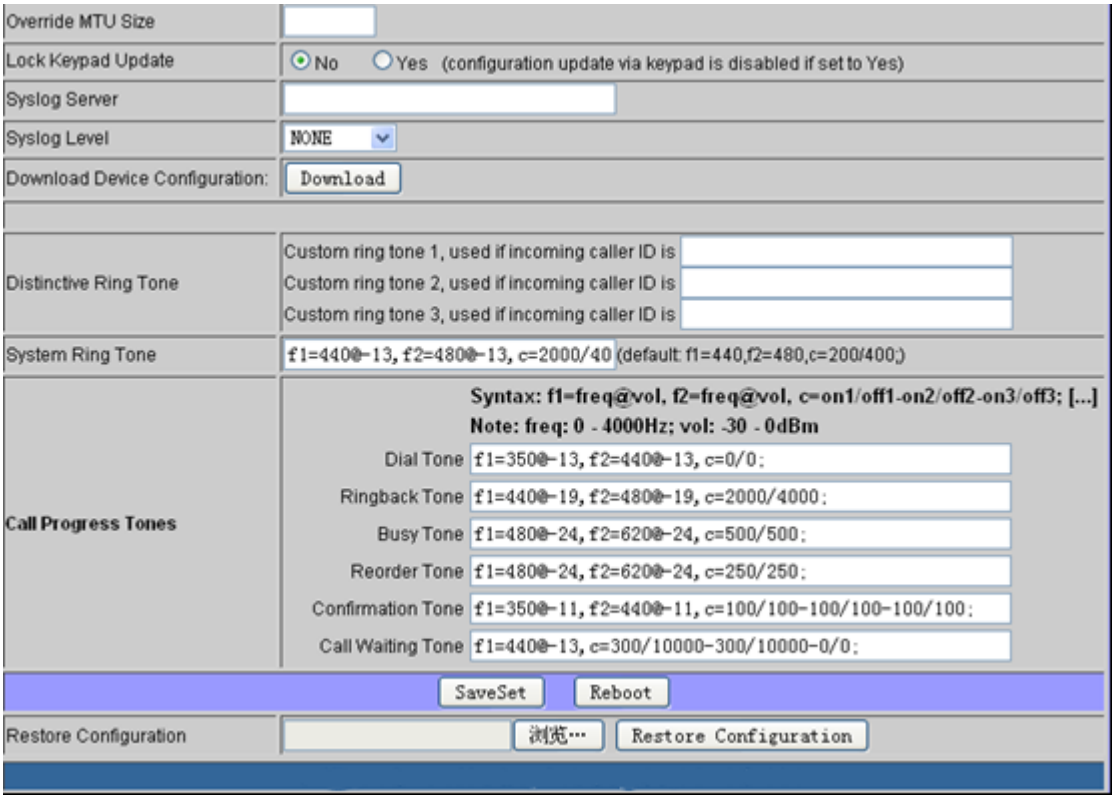

Super Options	
Setting Options	
Lock Keypad Update	If set to “Yes”, the configuration update via keypad is disabled.
Disable Voice Prompt	Default is No
Syslog Sever	The IP address or URL of System log server. This feature is especially useful for the ITSP (Internet Telephone Service Provider)
Syslog level	Select the GL-VP-150 to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog

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	payload, it contains the following components: GS_LOG: [device MAC address][error code] error message *Example*: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up
Download Device Configuration	Used to download the change you have made.
Distinctive Ring Tone	Caller ID must be configured. Select a Distinctive Ring Tone 1 through 3 for a particular Caller ID. The GL-VP-150 will ONLY use selected ring tones for particular Caller IDs. For all other calls, the GL-VP-150 will use System Ring Tone. When selected and no Caller ID is configured, the selected ring tone will be used for all incoming calls.
System Ring Tone	System ring tone. Default is North American standard. Adjust system ring tone frequencies and cadences based on local telecom standard.
Call Progress Tones	Using these settings, users can configure ring or tone frequencies based on parameters from local telecom. By default, they are set to North American standard. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. **Syntax**: f1=val,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]; (Frequencies are in Hz and cadence on and off are in 10ms) ON is the period of ringing (“On time” in ‘ms’) while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Up to three cadences are supported.
Restore Configuration	User can restore the before configuration from the configuration file saved at local pc

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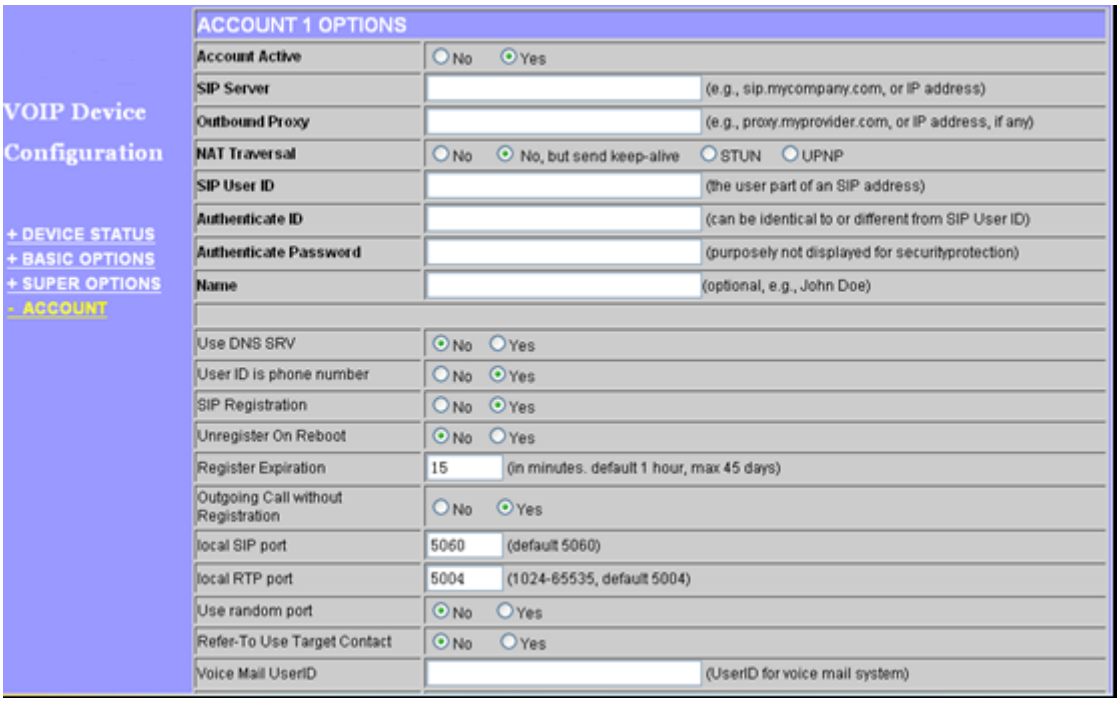
8.2.4.4 Account Options



Account 1 Options	
Settings Options	**Meaning**
Account active	When set to Yes the FXS port is activated
SIP Server	This field contains the URI string or the IP address (and port, if different from 5060) of the SIP proxy server. e.g., the following are some valid examples: sip.my-voip-provider.com, or sip:my-company-sip-server.com, or 192.168.1.200:5066
Outbound Proxy	This field contains the URI string or the IP address (and port, if different from 5060) of the outbound proxy. If there is no outbound proxy, this field SHOULD be left blank. If not blank, all outgoing requests will be sent to this outbound proxy.
NAT Traversal	This parameter defines whether or not the GL-VP-150 NAT traversal mechanism is activated. If activated (by choosing "Yes") and a STUN server is also specified, then the GL-VP-150 performs according to the STUN client specification. Using this mode, the embedded STUN client will detect if and what type of firewall/NAT is being used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the GL-VP-150 will use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" *with no specified STUN server*, the GL-VP-150 will periodically (every 20 seconds or so) send a blank UDP packet no payload data) to

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	the SIP server to keep the “hole” on the NAT open.
SIP User ID	SIP service subscriber’s User ID
Authenticate ID	SIP service subscriber’s Authenticate ID. Can be identical to or different from SIP User ID
Authenticate Password	SIP service subscriber’s account password
Name	SIP service subscriber’s name which will be used for Caller ID display
Use DNS SRV	Default is No. If set to Yes the client will use DNS SRV for server lookup
User ID is Phone Number	If the GL-VP-150 has an assigned PSTN telephone number, this field should be set to “Yes”. Otherwise, set it to “No”. If “Yes” is set, a “user=phone” parameter will be attached to the “From” header in SIP request
SIP Registration	This parameter controls whether the GL-VP-150 needs to send REGISTER messages to the proxy server. The default setting is “Yes”.
Unregister on Reboot	Default is “No.” If set to “Yes”, then the SIP user will be unregistered on reboot.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the GL-VP-150 refreshes its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Outgoing call without Registration	Default is **No**. If set to “Yes,” user can place outgoing calls even when not registered (if allowed by ITSP) but is unable to receive incoming calls.
Local SIP port	This parameter defines the local SIP port the GL-VP-150 will listen and transmit. The default value for FXS port is 5060. The default value for FXO port is 5062.
Local RTP port	This parameter defines the local RTP-RTCP port pair the GL-VP-150 will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP. The default value for FXS port is 5004. The default value for FXO port is 5008.
Use Random Port	This parameter, when set to Yes, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple GL-VP-150 are behind the same NAT.
Refer-To Use Target Contact	Default is NO. If set to YES, then for Attended Transfer, the “Refer-To” header uses the transferred target’s Contact header

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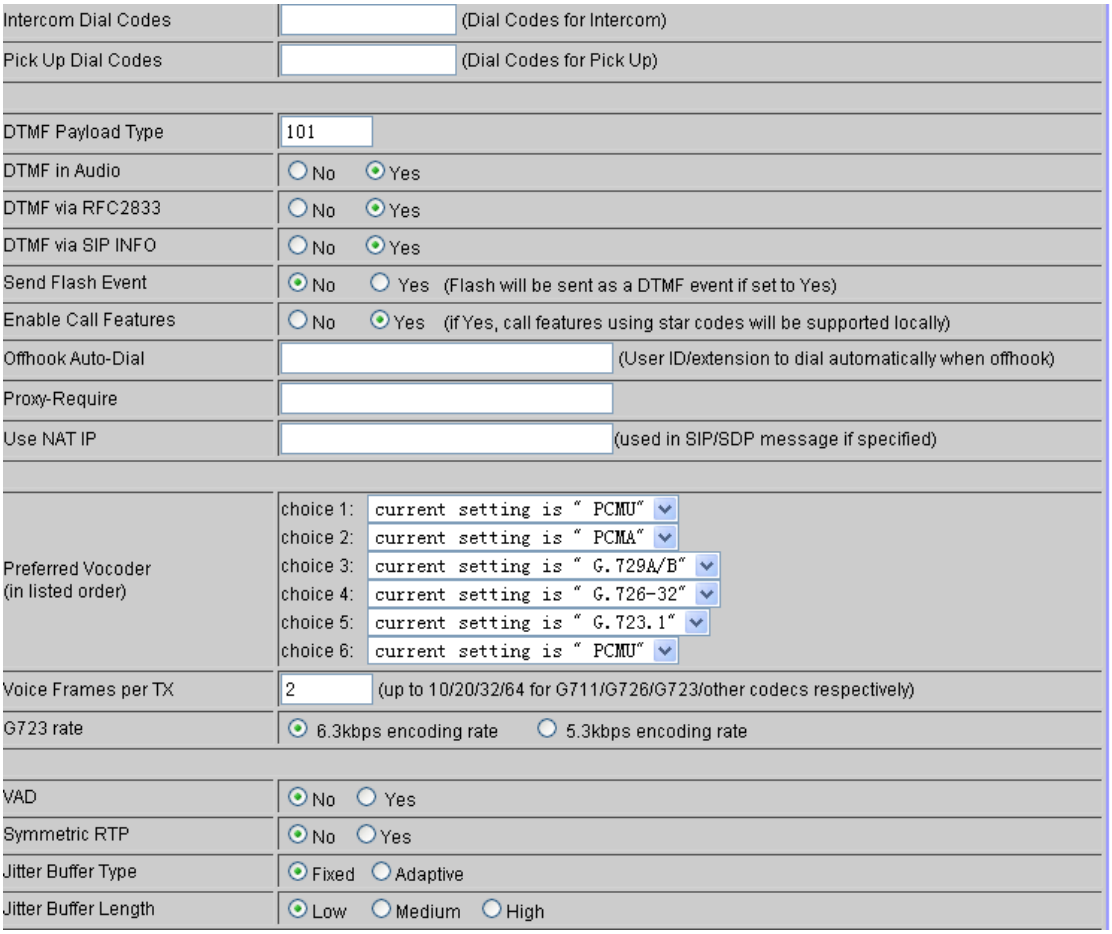
	information.


Intercom Dial Codes
(Dial Codes for Intercom)
Pick Up Dial Codes
(Dial Codes for Pick Up)
DTMF Payload Type
101
DTMF in Audio
No Yes
DTMF via RFC2833
No Yes
DTMF via SIP INFO
No Yes
Send Flash Event
No Yes (Flash will be sent as a DTMF event if set to Yes)
Enable Call Features
No Yes (if Yes, call features using star codes will be supported locally)
Offhook Auto-Dial
(User ID/extension to dial automatically when offhook)
Proxy-Require
Use NAT IP
(used in SIP/SDP message if specified)
Preferred Vocoder (in listed order)
choice 1: current setting is " PCMU"
choice 2: current setting is " PCMA"
choice 3: current setting is " G.729A/B"
choice 4: current setting is " G.726-32"
choice 5: current setting is " G.723.1"
choice 6: current setting is " PCMU"
Voice Frames per TX
2
(up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
G723 rate
6.3kbps encoding rate
5.3kbps encoding rate
VAD
No Yes
Symmetric RTP
No Yes
Jitter Buffer Type
Fixed Adaptive
Jitter Buffer Length
Low Medium High

Account 1 Options	
Settings Options	**Meaning**
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833
DTMF in Audio	Send DTMF as inband (in-audio).
DTMF via RFC2833	Send DTMF via RTP (According to RFC 2833).
DTMF via SIP INFO	Send DTMF via SIP INFO message.
Send Flash Event	This parameter allows users to control whether to send an SIP NOTIFY message indicating the Flash event, or just to switch to the voice channel when users press the Flash key.
Enable Call Features	Default is No. If set to Yes, Call Forwarding & Do-Not-Disturb are supported locally
Offhook Auto-Dial	This parameter allows users to configure a User ID or extension number to be automatically dialed upon offhook. Please note that only the user part of a SIP address needs to be entered here. The GL-VP-150 will automatically append the “@” and the host portion of the corresponding SIP address.

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Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
USE NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
Disable call-waiting	Default is **No**.
Preferred Vocoder	The GL-VP-150 supports up to 7 different Vocoder types including G.711 A-/U-law，G.723.1, G.726, G.728, G.729A/B, iLBC. Depending on the product model, some of these Vocoders may not be provided in standard release. Users can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder in this list can be entered by choosing the appropriate option in “Choice 1”. Similarly, the last Vocoder in this list can be entered by choosing the appropriate option in “Choice 7”.
Voice Frames per TX	This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. e.g., if the first vocoder is configured as G723 and the “Voice Frames per TX” is set to be 2, then the “ptime” value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first vocoder chosen is G729 or G711 or G726, then the “ptime” value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the GL-VP-150 will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.
G723 Rate	This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.

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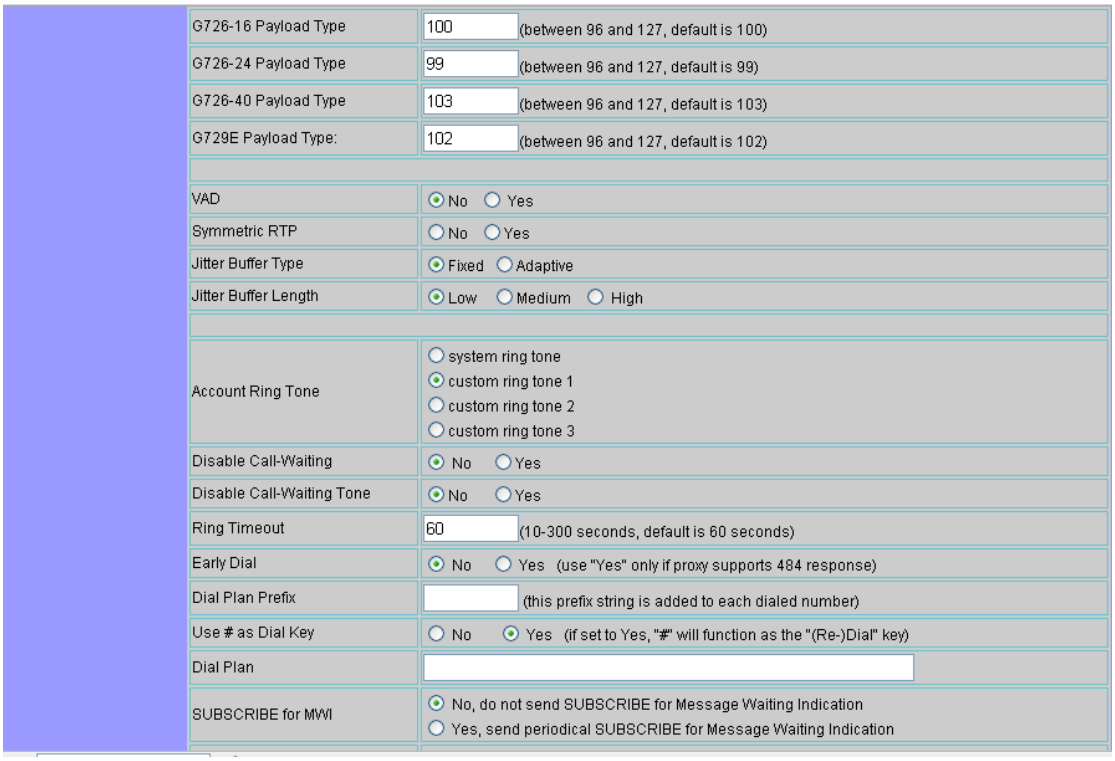

Account 1 Options	
Settings Options	**Meaning**
G726-16 Payload Type	Default value is 98. Range is from 96 to 127.
G726-24 Payload Type	Default value is 99. Range is from 96 to 127.
G726-40 Payload Type	Default value is 103. Range is from 96 to 127.
G729E Payload Type	Default value is 102. Range is from 96 to 127.
VAD	Default is **No**. VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.
Symmetric RTP	Default is **No**. When set to Yes the device will change the destination to send RTP packets to the source IP address and port of the inbound RTP packet last received by the device.
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions.
Jitter Buffer Length	Select Low, Medium or High based on network conditions.
Account Ring Tone	There are **4** uniquely defined ring tones: • One (1) System Ring Tone: when selected, all calls will ring with system ring tone. • Three (3) Customer Ring Tones: when selected, incoming calls

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	from designated account will play selected ring tone.
Disable Call -Waiting	Default is No.
Disable Call –Waiting Tone	Default is **No**. This is to disable the stutter Call Waiting Tone when a Call Waiting call arrives. The CWCID will still be displayed.
Ring Timeout	Incoming call will stop ringing when not picked up given a specific period of time.
Early Dial	Default is No. Use only if proxy supports 484 response
Dial Plan Prefix	Sets the prefix added to each dialed number
Use # as Send Key	This parameter allows users to configure the "#" key to be used as the "Send" (or "Dial") key. If set to "Yes", pressing this key will immediately trigger the sending of dialed string collected so far. In this case, this key is essentially equivalent to the "(Re)Dial" key. If set to "No", this "#" key will then be included as part of the dial string to be sent out.
Dial Plan	**Dial Plan Rules**:- 1. Accept Digits: 1,2,3,4,5,6,7,8,9,0 2. Grammar: x - any digit from 0-9; a. xx+ - at least 2 digit number; b. ^ - exclude; c. [3-5] - any digit of 3, 4, or 5; d. [147] - any digit 1, 4, or 7; e. <2=011> - replace digit 2 with 011 when dialing • **Example 1:** {[369]11 | 1617xxxxxxx} – Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617 • **Example 2:** {^1900x+ | <=1617>xxxxxxx} – Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers • **Example 3:** {1xxx[2-9]xxxxxx | <2=011>x+} – Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; If leading digit is 2, replace leading digit 2 with 011 before dialing 3. Default: Outgoing - {x+}
Subscribe for MWI	Default is No. When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.

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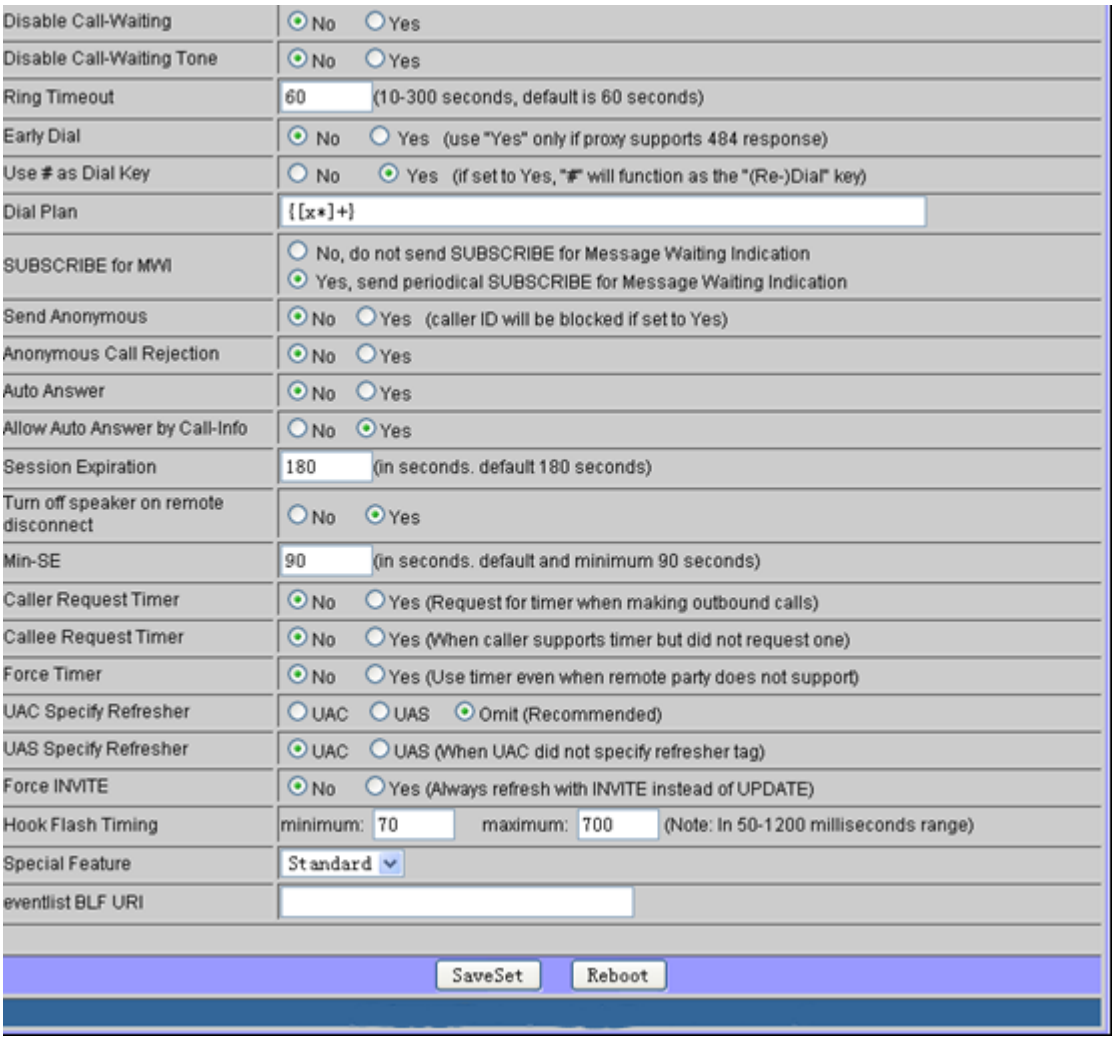
Disable Call-Waiting
No Yes
Disable Call-Waiting Tone
No Yes
Ring Timeout
60 (10-300 seconds, default is 60 seconds)
Early Dial
No Yes (use "Yes" only if proxy supports 484 response)
Use # as Dial Key
No Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
Dial Plan
{[x*]+}
SUBSCRIBE for MWI
No, do not send SUBSCRIBE for Message Waiting Indication
Yes, send periodical SUBSCRIBE for Message Waiting Indication
Send Anonymous
No Yes (caller ID will be blocked if set to Yes)
Anonymous Call Rejection
No Yes
Auto Answer
No Yes
Allow Auto Answer by Call-Info
No Yes
Session Expiration
180 (in seconds. default 180 seconds)
Turn off speaker on remote disconnect
No Yes
Min-SE
90 (in seconds. default and minimum 90 seconds)
Caller Request Timer
No Yes (Request for timer when making outbound calls)
Callee Request Timer
No Yes (When caller supports timer but did not request one)
Force Timer
No Yes (Use timer even when remote party does not support)
UAC Specify Refresher
UAC UAS Omit (Recommended)
UAS Specify Refresher
UAC UAS (When UAC did not specify refresher tag)
Force INVITE
No Yes (Always refresh with INVITE instead of UPDATE)
Hook Flash Timing
minimum: 70 maximum: 700 (Note: In 50-1200 milliseconds range)
Special Feature
Standard
eventlist BLF URI
SaveSet
Reboot

Account 1 Options	
Settings Options	**Meaning**
Send Anonymous	If this parameter is set to “Yes”, the “From” header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 486 Busy message.
Sesstion Expiration	The SIP Session Timer extension enables SIP sessions to be periodically “refreshed” via a SIP request (UPDATE, or re-INVITE. Once the session intervaexpires, if there is no refresh via a UPDATE or re-INVITE message, the session is terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, provided no successful session refresh transaction occurs beforehand. The default value is 180 seconds.

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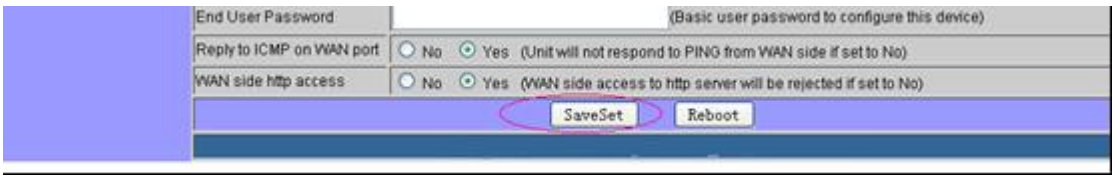
Min-SE	Defines the minimum session expiration (in seconds). Default is **90 seconds**.
Caller Request Timer	If set to “Yes”, the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting “Yes”, the phone will use session timer when it receives inbound calls with session timer request.
Force Timer	If set to “Yes”, the phone will use session timer even if the remote party does not support this feature. If set to “No”, the session timer is enabled only when the remote party supports this feature. To turn off Session Timer, select “No” for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
Force Invite	Session Timer can be refreshed using INVITE method or UPDATE method. Select “Yes” to use INVITE method to refresh the session timer.
Special Feature	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Lucent FS5000 Simple Endpoint, CBCOM, etc.

8.2.5. Saving the Configuration Changes

Once a change is made, users should click on the “SaveSet” button in the Configuration page, as follow:


End User Password
(Basic user password to configure this device)
Reply to ICMP on WAN port
No
Yes
(Unit will not respond to PING from WAN side if set to No)
WAN side http access
No
Yes
(WAN side access to http server will be rejected if set to No)
SaveSet
Reboot

The GL-VP-150 will then display the following screen to confirm that the changes have been saved. Please allow 5 to 10 seconds before rebooting the device.

VOIP Device Configuration
Reboot in progress...
You may login after 30 seconds by clicking the link below.
Click to relogin

8.2.6. Rebooting the GL-VP-150

You can reboot the GL-VP-150 by clicking on the “Reboot” button after each update to the configuration page. Alternatively, you can reboot by unplugging the power supply of the GL-VP-150 and then powering it on again. If your GL-VP-150 ever becomes “stuck” or un-responsive, you can unplug the power supply to reboot it. Frequent rebooting by unplugging the power supply is not recommended and should not be necessary.

8.2.7. Configuration through a Central Server

GL-VP-150 devices can be automatically configured from a central provisioning system.

When GL-VP-150 boots up, it will send TFTP or HTTP request to download configuration files. There are two configuration files, one is “cfg.txt” and the other is“cfg001fc1xxxxxx”, where “001fc1xxxxxx” is the MAC address of the GL-VP-150.

For more information regarding configuration file format, please refer to the related technical documentation.

The configuration file can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of GL-VP-150s can easily manage the configuration and service provisioning of individual devices remotely and automatically from a central server. The central provisioning system uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual GL-VP-150 for firmware upgrade, etc.

9. SOFTWARE UPGRADE

To upgrade software, GL-VP-150 can be configured with a TFTP server where the new code image is located. The TFTP upgrade can work in either static IP or DHCP mode using private or public IP address. It is recommended to set the TFTP server address in either a public IP address or on the same LAN with the GL-VP-150.

There are two ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt or via the GL-VP-150 ’s Web configuration interface. To configure the TFTP server via voice prompt, follow section 8.1, once set up the TFTP IP address, power cycle the GL-VP-150, the firmware will be fetched once the GL-VP-150 boots up.

To configure the TFTP server via the Web configuration interface, open up your browser to point at the IP address of the GL-VP-150. Input the admin password to enter the configuration screen. From there, enter the TFTP server address in the designated field towards the bottom of the configuration screen.

Once the TFTP server is configured, please power cycle the GL-VP-150.

TFTP process may take as long as 1 to 2 minutes over the Internet, or just 20+ seconds if it is performed on a LAN. Users are recommended to conduct TFTP upgrade in a

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controlled LAN environment if possible. For those who do not have a local TFTP server, GigaLink provides a NAT-friendly TFTP server on the public Internet for firmware upgrade. Please check the Service section of GigaLink's Web site to obtain this TFTP server's IP address.

NOTES:

When GigaLink IP Phone boot up, it will send TFTP or HTTP request to download configuration files, there are two configuration files, one is "cfg.txt" and the other is "cfg001fc1xxxxxx", where "001fc1xxxxxx" is the MAC address of the GL-VP-150 . These two files are for initial automatically provisioning purpose only, for normal TFTP or HTTP firmware upgrade, the following error messages in a TFTP or HTTP server log can be ignored.

10. RESTORE TO FACTORY DEFAULT SETTINGS

Warning:

Restoring to the factory default settings will delete all configuration information of the device.

Steps to follow in restoring to factory default settings:

Step1: Return to the Config sub-menu.

Step2: Choose the Factory Reset sub-menu and press **MENU** again.

Step3:Choose Confirm Factory Reset or Cancel Factory Reset.

11. TECHNICAL SUPPORT CONTACT

Email: info@gigo-link.ru

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